DEVICE
the newsletter for the electronic guitarist/musician VOL.1:3-79

DEVICE INTERVIEW:

WHO IS STEVEN ST. CROIX AND WHY IS HE SAYING THOSE THINGS ABOUT DELAY LINES?

This is the first part of an interview with designer/innovator Steven St. Croix, V.P. in charge of Research and Development for Marshall Electronics, 1205 York Rd. Suite 14, Luterville, MD 21093 (not to be confused with the amplifier manufacturer). Our interview was conducted in late '78 by our East Coast correspondent, Pete Johnson, at Steve's recently completed 24 track studio tucked away in the rolling hills outside of Baltimore. A man of many talents, Steve leads a busy life apart from his work with Marshall. A working musician, he has done extensive studio work with, among others, Stevie Wonder. His current projects include completing work on albums of his own, the ultimate in-car sound system, audio for a new film, and the rebuilding of various sports cars. His other passions include scuba diving and Chinese food. He also claims to have the fastest car on the road.

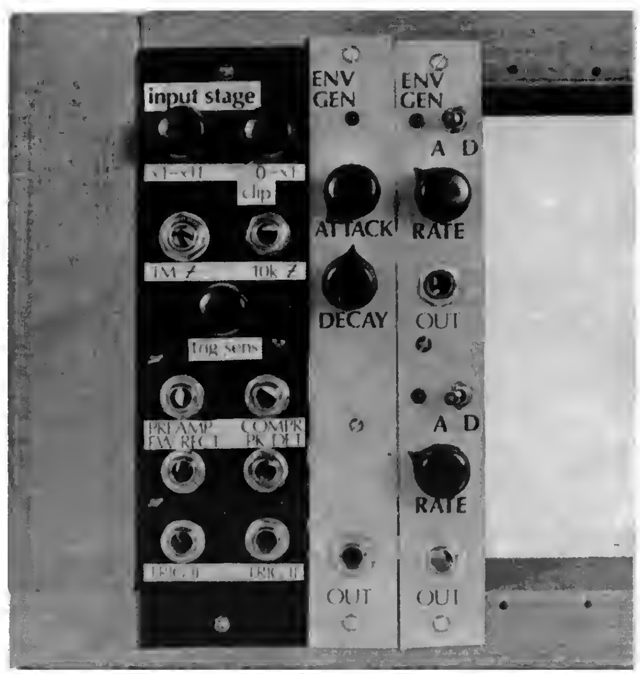
Pete: What's your background, and how did you get into the electronic music field?
Steve: Well, about 14 or 15 years ago I was living in Europe and playing rock and roll. I was working with some people in England trying to establish ways to get new sounds, but not new sounds as in playing convoluted chord changes; new sounds as in new audio effects. At that time there just weren't any special effects devices happening in music. A lot of people had Echoplexes, and that was great...but then people started finding out that if you played real loud, it sounded better than if you played soft and that the overall (*Cont. on page 8*)

IN THIS ISSUE:

construction: BUILDING THE AMS-100 - part 3
by craig anderton

POWERING AND PACKAGING: Before getting into next month's VCA module, I thought it might be a good idea to discuss packaging and power supply considerations for the AMS-100 modules; let's begin with packaging.

Figure 1 (below) shows the input module from DEVICE #1:1 along with the envelope generators from DEVICE #1:2, built up (*Cont. on page 11*)



REVIEW:
THE EH MICRO SYNTHESIZER

...or, how much synthesis can you expect for under $200? Read the following and find out.

The Electro-Harmonix Micro-Synthesizer is a small (8" x 6½" x 1½") effects box designed to, in E-H's words, "create many of today's most popular lead synthesizer voicings at a fraction of the normal cost for such capabilities". It does not contain a pitch-to-voltage converter or internal oscillators, but is rather a collection of processing units put together in a single floor-mounting package, with an in/out footswitch and built-in AC power supply.

The box is attractively and logically laid out, with 10 slide pots across the front and the footswitch in the lower left-hand corner; however, the panel graphics may be easily scratched, and the front would probably get that beat-up look pretty fast in a live performance context. (Tip: if you have a Micro-Synthesizer, carefully take it apart and, equally carefully, cover the top with clear contact covering. Then cut holes for the sliders with an X-Acto knife or equivalent...this will help the paint stand up to the test of time.)

Functionally speaking, the effect is designed to accept a standard level, non-preamped guitar output, although there is a preamp trimpot access hole in the bottom to trim the preamp gain to an optimum level. The input impedance is about 110 KOhms.

After the preamp, the signal splits into 4 voices: normal guitar, octave higher, octave lower (suboctave), and distortion (fuzz). Each of the voice outputs connects to a slide pot, and all outputs are mixable in any desired proportion. Another interesting feature is that the voices are dynamically responsive; that is, the harder you pick, the louder the volume. The correlation is not perfect, but comes very close. There also seems to be a type of noise gating effect that cuts off the signal when it goes below a certain level. I don't think this is a noise gate per se; my guess is that it's some type of hysteresis on the squaring stage that drives the octave/suboctave sections. In any event, the result is similar.

The guitar voice is clean, and is filtered so that response starts dropping off rapidly above 10 KHz. The octave above voice is a "fuzzy" type of octave sound that is most useful in the upper range of the guitar, and sounds like the full-wave rectified fuzz type of circuit we're using in the AMS-100. The response of this stage is peaked in the 300 Hz to 1 KHz region. While this no doubt increases the illusion of the octave above effect, it also adds a "boxy" character to the sound that doesn't really knock me out. Perhaps the filter could be changed to a high pass instead of bandpass type to give the same results, but with a little more open sound.

The suboctave is also a bit dirty, although the dirtiness is an advantage here; it gives a bigger effect that helps the bass cut through more, and also makes for more interesting filtering effects. One thing that mars the sound is some high frequency buzzing on the signal; however, it sounds like this might be feed-through generated from another stage instead of being an integral part of the signal...as a result, there may be some modification that would eliminate the problem. The frequency response of this stage is weighted towards the bass side, with the response down to about half amplitude at 1 KHz compared to 50 Hz.

The distortion voice produces a square wave at the output. It will not respond predictably to signals above 4 KHz.

An important question is how well the octave above, and below, track the input signal. The answer is, quite well. As with other systems of this type, using the bass pickup gives the best results, but the treble and both pickup positions also do the job (albeit not as well). However, it can be fooled into producing real garbage if your pickups are put out of phase with each other, or if you play two notes at once. I still haven't heard an octave divider that was perfect, but the one in here is musically usable at the very least.

After setting up the desired mix of the four voices, the combined output passes through a VCA hooked up as an attack delay unit. The triggering characteristic requires you to put a space between notes for the triggering circuitry to reset; if you observe that rule and play relatively cleanly, the thing will trigger reliably... but I do have one strong reservation about this, which we'll get into later. The attack time of the VCA is adjustable, but the

decay time is not.

From here, the signal moves into a low pass filter that sweeps from below 50 Hz to 9 KHz at the top end, with variable resonance. Actually, the filter is quite novel, so I'd like to get into it in some detail.

The filter has 4 controls, arranged in the following order on the panel: resonance, start point, stop point, and rate. The start point specifies where the filter will start its sweep, while the stop point specifies at what frequency the filter sweep ends. The rate control determines how long it takes for the signal to get from the start point to the stop point. As a result, if the start point is set lower than the stop point, the filter will sweep upwards; if the start point is set higher than the stop point, the filter will sweep downwards. By putting the start and stop controls in the same position, no sweeping occurs and you can achieve static filter effects. There is an additional trigger control for the filter that triggers a sweep whenever your playing exceeds the level specified by this control. Therefore with a less sensitive trigger adjustment you will trigger on every note (and re-trigger when you hit chords, whether you want to or not). As a result, there is a bit of compromise involved in the setting of this control...senstive enough to be responsive, but not so sensitive that it constantly re-triggers when not wanted.

The in/out switch does not bypass the signal entirely, and the output phase is reversed compared to the phase of the input signal. This can be a disadvantage if you wish to parallel this box with other boxes, although of course E-H never intended the Micro-Synthesizer to be used in this manner.

EVALUATION. I had a lot of problems getting the unit to behave properly when I first tried it out, so much so that I just about dismissed it as unusable. However, this was due to an incorrect setting of the preamp trimpot, which is crucial to optimum performance. Unfortunately, this did not completely solve the problem. When the trimpot is adjusted as specified by E-H (for minimum distortion on peaks), the signal overloaded the VCA trigger and caused severe false triggering and retriggering problems. Backing off on the trimpot produced an excellent triggering characteristic, but lowered the overall amplitude of the signal so much that the signal-to-noise ratio suffered. I also noted that when the unit was completely bypassed (via external switching), the amplitude was considerably lower than the original input signal. Now, either this is a production flaw like a mis-installed part, or there is a glitch in the design. Whether my particular unit is "up to spec" or not, I feel E-H should definitely put in a separate trigger control for the attack delay section. There are just too many different playing styles and picking techniques to expect a single preamp adjustment to also put things in the right range for the trigger; for example, playing with a thumbpick or a flat pick produces an entirely different kind of attack transient which cannot be compensated for with the Micro-Synthesizer, even though the <u>apparent</u> level of the signal is still the same.

I really have no other complaints, although I should address the problem of reliability. I don't think I'm letting out any secrets when I say that many musicians are frankly skeptical of the reliability of E-H units; however, this must be put in perspective with the cost of the unit. For example, the corners of the case are not welded, meaning that the corners can get bent out of shape fairly easily; the exposed linear slide pots just beckon for dust to come in (and E-H isn't the only company that does this); there are six little screws holding the bottom plate in place...and they must be in place for the unit to have any structural integrity; the slider "knobs" could easily break off to one side or the other; and as mentioned earlier, the graphics don't hold up well to scratching. But let's face it. If E-H used studio quality faders, cold rolled steel cases with recessed knob positions and all the rest, the thing would sell for some huge amount of bucks and no one would buy it. I'll be using the Micro-Synthesizer in the studio, and I expect it to last a long time simply because it will not get that on-the-road abuse. This is a unit that can tolerate use, but I think it would be pretty unforgiving of abuse.

CONCLUSION. I think E-H has done what they set out to do; the box does imitate synthesizer voicings, expand the sounds you can get with a guitar, and brings it in for a low price (it shows up in music stores for around $180). The choice of voices and options for the unit are clever; I particularly like the simplicity of the filter, and the fact that if offers a lot of flexibility for having only 4 controls. The octave above sound is adequate, the suboctave sound is quite good, and when you mix the

(Continued on page 4)

(*Micro-Synthesizer cont. from page 3*)
various voices together you can indeed get some neat sounds...especially once you know your way around the filter, and have acquired a playing style that is compatible with the machine's triggering mechanism. All in all, I think it's a pity that a unit this versatile doesn't have some way to adjust the attack delay trigger, because the lack of this control can fool you into thinking the unit doesn't work as well as it can. Keeping the signal up makes the octave divider/multiplier functions happy, while keeping the signal down gives better triggering...it would be nice to be able to adjust both independently. I should add that I do pick harder than most people, so perhaps other players would not have this problem; but then again, I feel hard pickers should not be excluded from using this box.

If you're expecting the Micro-Synthesizer to be a micro-Avatar, you're going to be disappointed; the M-S is more like some kind of maxi, highly sophisticated fuzz box with a built-in wide range, intelligent wa pedal. That may not appeal to players who have the technical chops to build an AMS-100 or the bucks to afford something like the BCD "Nebula", but it could very easily be the answer to a prayer for a club musician who wants to get synthesized sounding licks without paying full synthesizer prices. In the studio, it's not as versatile a unit as a delay line, but it definitely has its moments...with guitar it works great for brass-like timbres and multiple melody line overdubs, and it also seems to have a real fondness for the PAIA Programmable Drums. Now, all we need is someone from E-H to write in with a modification on how to add a separate triggering control to the attack delay (or even how to use the filter trigger slider to control both the VCA and VCF)...I'm all ears. Or if not that, how about a schematic so I can figure out a modification myself?

---*Craig Anderton*

REVIEW:

SCHECTER 'TELE' PICK-UPS

I'm sure you've seen the proliferation of "hot pickup" ads over the last few years; and like me, you've probably been wondering if they're worth the bucks and trouble. Recently, I got ahold of Schecter's tapped Telecaster replacement pickups (stock numbers: lead pickup, F520T; rhythm pickup, F521T) in order to see if I could get a little more variety in my guitar sound; here are the results of playing with these pickups.

COMPATIBILITY WITH THE TELECASTER. One question with replacement parts is whether they really are compatible with the instrument...do you have to do additional routing, fight with screw holes that are misaligned,

[*Are hot rod pickups worth it? Here's what happened when Craig decided to upgrade his Telecaster.*]

and so on. My Tele is a 1966 model, and with the exception of a couple of points we will examine later, the pickups were easy to mount in place and fully compatible with the guitar. No sanding or messing around was required to make things fit right. By the way, when you mount the treble pickup you'll have to unscrew the bridge assembly. When you put the guitar back together again, make sure that the bridge is screwed down tightly (but obviously, don't strip the wood). This is important to keep the guitar as solid an overall unit as possible.

HOW ARE REPLACEMENT PICKUPS DIFFERENT? These replacement pickups have both more coil winding and more magnet power, which increases the output and modifies the tonal quality. Other changes relative to the stock Tele pickups are a sturdier feel and frame, a center tap on the coil to give a lower output sound that is closer to "stock"

DEVICE notes

Effective in APRIL, DEVICE's operations are moving to Austin, Texas. Roger has been offered an attractive job there with Unicord; since he is most involved with the day-to-day running of DEVICE, it seemed natural that the business follow him around. This in no way diminishes my involvement. In fact, since Roger will no longer do reviews to avoid conflicts of interest, we will be carrying more material by myself and others.

Until we announce the new address in DEVICE, address all inquiries to the Carmichael address as they will be forwarded. Due to the move, we are anticipating a 10 day delay for the April issue. ---Craig

pickups, and a copper shield - grounded to one of the terminals - that extends around the windings for shielding. Overall, I'd say the pickups are well made...wax covered windings and the like; they definitely don't have a cheap feel about them, and are much more substantial than the stock pickups.

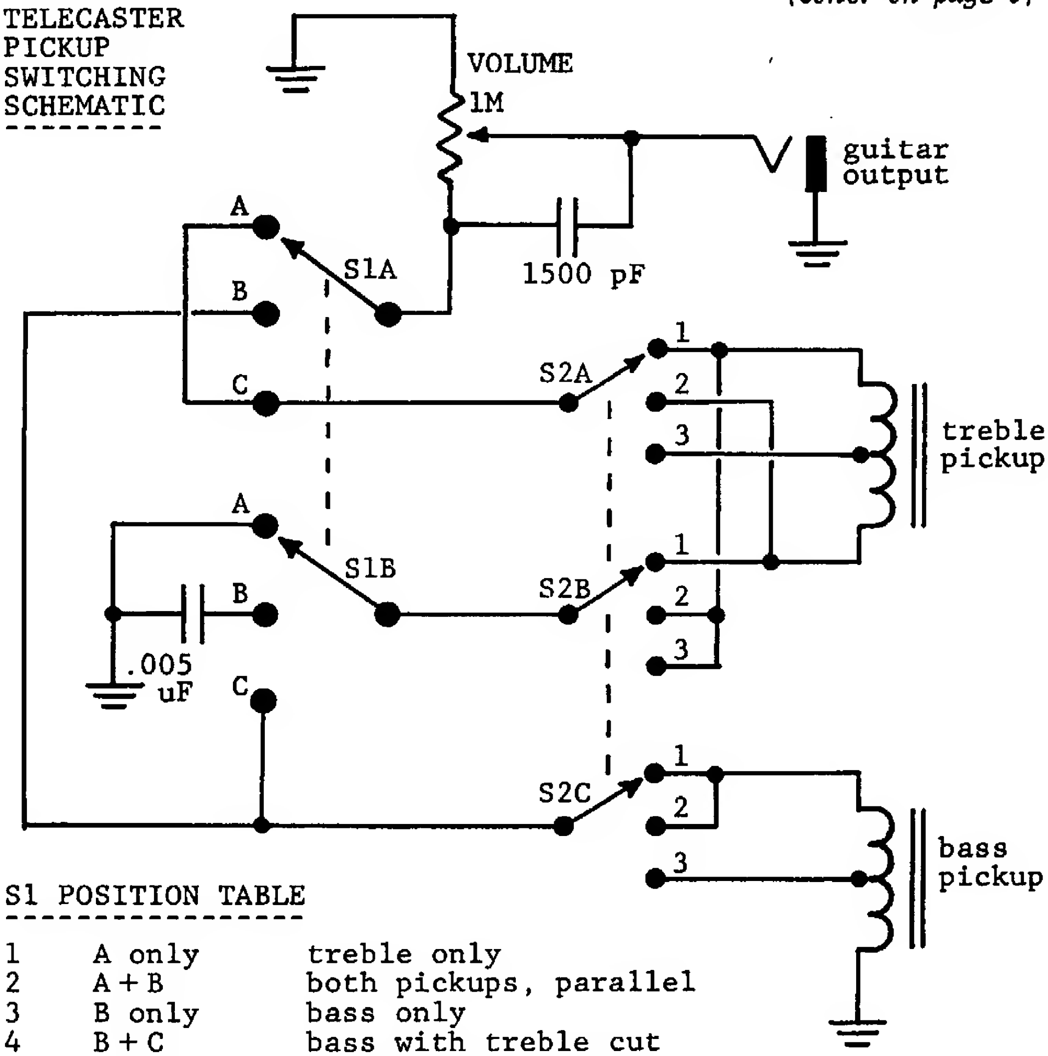
The Schecter catalog specifies both pickups as 16,000 Ohm units, with a tap at the 8000 Ohm point. However, neither pickup measured up to this spec as noted in the table below (readings taken with Sinclair DVM, pickups out of circuit):

pickup	*total DC Ohms*	*tap point*
Bass	11,710Ω	6,160Ω
Treble	13,790Ω	7,330Ω

Both tap points were quite close to center; the lead pickup exhibited a 400 Ohm variation from exact center for an error of only 3% compared to the total winding. The bass pickup is about 300 Ohms. Unfortunately, it should be noted that DC resistance readings are not always accurate for determining the number of turns or AC impedance, but they're what we have to go with at the

(Cont. on page 6)

TELECASTER PICKUP SWITCHING SCHEMATIC



S1 POSITION TABLE

1	A only	treble only
2	A + B	both pickups, parallel
3	B only	bass only
4	B + C	bass with treble cut
5	C only	both pickups, series

S2 POSITION TABLE

1	Treble pickup in phase with bass pickup
2	Treble pickup out of phase with bass pickup
3	Both pickups tapped at halfway point, with treble pickup out of phase

(*Schecter pickups cont. from page 5*)
present. Incidentally, Schecter does not specify whether the Ohms figure in the catalog is for DC resistance or AC impedance.

ACCOMPANYING ELECTRONICS. I had already modified my Tele once before (see the May '78 issue of Guitar Player for details), but didn't like using toggle switches and the non-original electronics plate. There are ways to do a more flexible wiring if you wish, which we'll mention at the end of this section. Note that in order to accomplish this wiring, you need to remove the jumper on the underside of the treble pickup (it connects a terminal to the ground plate/bridge assembly). Make sure you connect a wire to this ground plate assembly and connect it to the ground terminal of the output jack. The other three wires are the pickup wires shown in accompanying schematic.

S1 is the pickup selector switch, and is a 5 position Schecter replacement switch. While usually thought of as a replacement switch for Stratocasters, it also works well with the Tele and does not require any mechanical modifications. In position 1, treble pickup only, the wiper connects to contact A only; in position 2, both pickups in parallel, the wiper connects to both contacts A and B; in position 3, bass pickup only, the wiper connects to contact B only; in position 4, bass with treble cut, the wiper connects to both B and C; and in position 5, both pickups in series, the wiper connects to contact C only.

S2 is a 3P3T switch. In position 1, the treble pickup is in phase with the bass pickup; in position 2, the treble pickup is out of phase with the bass pickup; and in position 3, both pickups are tapped at the half-way point to get the "stock" sound, with the treble pickup out of phase to make things more interesting.

This setup gives 12 basic, and quite different, sounds. More sounds could be obtained by replacing S2 with three individual switches, one for the treble pickup phase reverse function, and the other two for providing independent means of changing the bass and treble pickups from full coil to half (tapped coil) modes. I hope to fool around with these pickups some more, using some of Schecter's pots with click switches on the back, to implement some even hairier wiring schemes.

EVALUATION OF SOUND, TAPPED COIL MODE. In the tapped mode, where you are using only half of the pickup winding, the sound is remarkably similar to the stock Tele sound, although there appears to be a bit more presence. Like a stock Tele, the out of phase parallel and series effects are thin-sounding, but useful on special occasions. It's interesting, though, that even with adding hot rod pickups, you can still achieve the familiar Tele sound; I love that particular sound, and didn't want to have to give it up for the sake of more output.

FULL COIL MODE. First of all, the individual pickups sound "bigger", with an apparent low end and lower midrange emphasis that creates a thicker, slightly darker sound with more output. The highs do not seem lessened; rather, the lows seem more prominent. As a result, the sound lacks some of the delicacy of the original Fender pickup...which sometimes is EXACTLY what you're looking for.

The series mode (treble in phase) gives out a big blorb of power, but the sound doesn't really get to me as it seems somewhat ill-defined and heavy sounding. It would probably be great for chording through Marshalls (in fact, I'm sure it would be) but one reason I like the Tele is its subtlety, so this isn't my favorite sound. However, if you have an effects box that requires a lot of drive, this is your baby.

The parallel mode (again, treble in phase) was a bit more under control, and sounded very nice...sort of that Fender surfing sound that was popular many years ago, except with a lot more punch and less thinness. But, it was the out of phase options that really surprised me. Due to the extra sock and output of the pickups, and also thanks to the extended low range, the out of phase sounds are not thin-sounding at all...they have a rich, yet distinctive, sound that is less powerful than some other full coil positions, but considerably more powerful than equivalent sounds using only half the coil.

COMPLAINTS. Yes, I do have some. The pole pieces of the replacement pickups, like the original Tele pickups, are not adjustable which causes different response from different strings. Also, the 1st and 6th strings lie closer to the face of the pickup, since the neck and bridge of the Tele are curved but the face of the pickup is straight; this can lead to problems with the low E string just putting too much juice into the pickup. My way of dealing with this problem was to screw the pickups well into the body and away from the strings so that the curvature had less ef-

fect, although this does not diminish the output somewhat. However, since these are pretty high output pickups, the slightly lower output can be easily tolerated for the sake of a more even response. By the way, when I had the strings up real close to the pickups there seemed to be a bit less sustain...perhaps the magnets are powerful enough to slow the strings down a bit. If so, that's another good reason to keep the pickups a bit further away from the strings than normal.

There was also a small problem with regards to mounting the bass pickup. With its slightly thicker bobbin, I couldn't get it far enough into the body for my tastes; the little springs that go between the pickup and the body would compress only so far and no further. However, I simply used the springs for the stock treble pickup (they were no longer needed since Schecter provides mounting hardware for the treble pickup) in place of the springs for the bass pickup, and everything worked out fine.

I also feel that some more documentation should have been included with the pickups so that users could be aware of the many possibilities inherent in a tapped coil design. As it is, you're pretty much on your own.

OVERALL EVALUATION. I'm happy to say that I am very pleased with the final results of modifying my guitar. What I've really gained is many more basic sounds - some with less output, some with more output, some with different response - yet, none of this requires batteries, complicated wiring, or excessive numbers of controls on the guitar. The Tele's sound is cleaner and definitely more substantial. Part of the reason why I'm happy with the modification is the tapped feature of the coils. While I like the hot rod sound, I'm glad I didn't have to give up the stock sound. My one trepidation at installing these pickups was that I would lose the sound I've grown accustomed to over the years - I didn't. Are hot rod pickups worth it? In my case, the answer is yes.

While this review has been Schecter-oriented, I'd like to emphasize that there are plenty of other companies out there making hot rod pickups; it's just hard to try them all out on an objective basis, so I settled on one manufacturer whose products have impressed me in the past. If you have compared hot rod pickups in your own guitar, let us know your opinions. --*Craig Anderton*

PRELUDE to PEDALBOARDS

roger clay

This series of articles is aimed mainly at those who are uninitiated in electronic construction. Our main purpose is to create an operating, rugged system for the myriad effects you've probably acquired over the years. I'm sure you are all familiar with the problems that surround the use of a bunch of devices: signal loss, miles of patch cords, bending down to adjust different knobs and switches, increased noise... and these are just the more noticeable hassles. Getting rid of them is theoretically a "simple" idea but one that only a few of us have actually tackled ourselves. For years I tried in vain to enlist some of my more technically inclined friends to help me put together something that would solve these problems, but they never seemed to have the time to devote to my "simple" project. This series is dedicated to those who found themselves in similar situations. At the same time, if you have built your own systems please don't go away. It's my hope that as we progress, you will contribute some of your findings and help the column evolve.

Before we start getting our hands dirty, some preliminary research of materials is necessary. There are many factors that go into design and construction, and understanding how to deal with the questions that will arise is of the utmost importance to us.

A basic design objective must be decided on first. In my case, what I wanted was a system that placed all the "fine tuning" controls within easy viewing range and at arm's length. My first attempt at something along these lines was with an old E-H "Big Muff". By removing the footswitch from the Muff's chassis and mounting it in a discarded case with a long cord of shielded cable I had the volume, tone, and sustain controls at my fingertips. It was funky and intermittent, but it was the basis for formulating a future objective. So, carefully examine the motions you make when you use an effect, especially during live performance. Notice the extra maneuvers you make and take note of how and where things can be simplified. This may

(Cont. on page 13)

STEVEN ST. CROIX

(*Cont.*) sound had a little more balls. I was working with someone from Vox at that time, they were just coming out with the "Super Beatle" amps, so that tells you what time it was. Anyway, there were some people playing in the area I was in who were trying to get something a little more aggressive-sounding, and we had this tape recording that was brought to us by Jeff Beck. He told us he wanted his guitar to sound like it did on the tape; what he showed us was one of those old tape machines with the magic eye level indicators and he said, "if the eye is closed all the time, full on, it sounds great". What he was doing was putting the output of a preamp into the mic input and blowing the Hell out of the front end of the machine. It was, of course, distorting. From that came the Arbiter "Fuzz Face" and the Vox "Mini Distorter"...but the "Fuzz Face" was the one that took off. That's how it all started: in Europe, with effects that gave you a little more sustain and a little more balls on stage.

Because I was a working musician, I could tell right away if an effect was musically useful or not, while a lot of other people who were doing engineering were engineers only. They could come up with all kinds of hot stuff...but it was stuff that didn't necessarily apply to music. This is still a problem today; you've got musicians and engineers in a lot of companies, but they're usually not the same people. The engineer may design this really beautiful thing but in a real performance situation it has little musical value...you'll notice this a lot in console equalization modules. Anyway, our experimenting went on and we kept developing more toys. During all this I had been doing medical electronics, going to school in Switzerland, and playing music full time. It was a pretty packed period of time, so it didn't really occur to me to do gizmo manufacturing. I was doing designing and design licensing for other people, and that seemed enough. It stayed that way until about 4 years ago.

P: Was that about the time you developed the Marshall Time Modulator?

"(ENGINEERS) COULD COME UP WITH ALL KINDS OF HOT STUFF... BUT IT WAS STUFF THAT DIDN'T NECESSARILY APPLY TO MUSIC. THIS IS STILL A PROBLEM TODAY; YOU'VE GOT MUSICIANS AND ENGINEERS IN A LOT OF COMPANIES, BUT THEY'RE USUALLY NOT THE SAME PEOPLE."

S: Yeah. Remember, this was before there were any flexible time-based manipulators. A friend and I were having a discussion about manipulating time and the overall ramifications as far as communications was concerned. The conclusion we came to was that it would be a lot of fun to figure out how to do audio time modulation and hear the results. I had four days before I was due to go out on a gig, so that night I sat down and was determined to show my friend that I could back up all my outrageous statements about time modulation. I had been recently doing work with charge transfer systems, doing a lot of R & D, and I decided to incorporate this research in the design of this new idea. I stayed up all night, really cracked on it, and by lunch the next day I had a working prototype. It was literally an overnight device.

P: Could you explain "charge transfer systems"...

S: An analog system of clocked delay. I knew we had to go analog on this idea, and the only analog system that could be practical would be a clock transfer system with a FET (field effect transistor) and a capacitor. The cap would remember the voltage, and be switched to the next cap by a clock. We didn't want to take a digital approach, because the inherent limitations would have eliminated many of the possibilities I was heading for.

P: Is charge transfer similar to bucket brigade?

S: Well it's similar, but there were some differences.

P: Back to developing the prototype...

S: Well, the idea was to get a transfer system working, which I did. It was incredibly unwieldy and super-touchy, but it worked. The afternoon I'd gotten it working I had a session in the studio, so I took the device along to try on a song I was doing. There were a bunch of folks hanging around including the studio owner and I said "I've just got-

ten this thing working, you should listen to it". I'd never really tried the thing out with music, I'd just talked through it...I'd say "hello" and the thing would go "whuaguaula". I thought, "wow, this is great!" It had a dynamic range of about 1 dB and a bandpass of about 1 Hz, not real impressive in terms of performance! Conceptually though, it was a mind-fucker. I brought it into the studio and played with it a couple of minutes while I was working on a Clavinet part...those couple of minutes turned into hours. It wasn't intended to be a product; it was made to win a bet, quite frankly. But as we played with it, it dawned on us that this was an audio product that could really be useful.

In the meantime, the AES show was a couple of days away. Now I'd never even heard of the AES, let alone gone to an AES convention. But the guy who owned the studio suggested we make the thing look a little more realistic and take it to the show in Los Angeles, rent a booth, show it to some people, and see what happens. That seemed like an interesting idea, so I went home and went crazy getting the thing ready. Did a silk-screen front panel, added something like 8 times the number of delay elements to bring the bandpass up, and some other stuff. We packed the whole thing up in a paper box, we didn't even have a chassis; printed some real fast brochures at a jiffy overnight printer, rented the smallest booth we could find and walked in cold. I mean, I had hardly even heard of the AES and here I was showing at their convention. I had no fixed idea of what to do, I just wanted to show people the toy and see what they thought. It was still light years away from production, but it was an interesting concept. So we showed it...and all Hell broke loose. It just went crazy, the booth was constantly crowded. People were telling other people to come in and see it...it was packed! Well, we walked away with two years worth of orders. And it took those two years to fill 'em. We decided not to ship them until we had finished developing this machine into a truly professional device.

We got a lot of yelling and screaming for pulling the thing off the market during this period, but I had been looking at the marketing mistakes of other audio products and came to realize you have to make a basic decision with a new product---let it out like it is and try to make a whole bunch of money, or wait and make it absolutely the best you can and then let it out into the world. You may lose some money or customers at the start, but the theory is it will really do you good in the end because even your oldest products will be really excellent. We ran several hundred prototypes - the PT series - they worked great, but they were still prototypes. They were touchy and didn't have the Kepton flex circuits we use now; they didn't have the heavy weight metal and welded studs, but they were very similar in appearance and performance to the ones out now. Those were the one you'd call the original models; those were the ones Leo Kottke and Stevie Wonder got.

P: How were they different from the ones out now?

S: They only had one delay line instead of two, they only had a sweep rate of 40:1 which is now 72:1, the dynamic range was only 85 dB - now we spec 95 dB unweighted at the longest delay. They didn't have custom pots, the metalwork wasn't strong, things like that...we've put a lot of effort into making a more roadworthy product.

P: What is the basic principle of the Time Modulator's operation?

S: There are two main families of ways to delay; digital and analog. Due to developments in the past year digital now falls into two basic categories: conventional digital, where the signal is converted by an analog to digital converter to a bunch of numbers. These numbers are then clocked down a series of delay lines. If it's, say, a 10 bit digital line then there are 10 parallel delay lines in the device, whether it be RAM (random access memory) or shift register. Each line passes a bit down the line in parallel. At the end, or in the middle if you want less delay, the line feeds a digital to analog converter and is translated back into analog. That's conventional old telemetry digital that they've been using forever. The advantages are: if you want to spend the money, you can get as much delay as you'd like without degradation of quality because digital is a regenerative process. If a bit of information starts to get "sloppy" it doesn't matter, because a bit is either "on" or "off"; the next time it's transferred to the next stage, it sharpens right up again. So there is no transfer loss in digital. There are dropped bits and mistakes, but there is no necessarily cumulative loss in digital, which is the big advantage of digital technology. If you want 5 seconds of delay, you just whip right

(Cont. on page 10)

(*Steven St. Croix cont. from page 9*)
out and buy yourself 5 seconds of digital delay line. But there are also disadvantages. Digital lines are expensive, they run hot, they can't be swept over a wide range, and it is a bitch to generate a lot of the special effects you can theoretically do in analog. And, digital lines sound digital; when you convert in and out of digital you only have so many numbers or steps. Remember, the digital line doesn't remember pitch, it remembers samples of amplitude...hopefully at least 15,000 samples per second to get a realistic picture of the waveform. To get an anywhere near accurate sample of a signal you need to have at least twice as many samples as the highest frequency, so you're talking about some pretty high clock rates. What actually happens is this: the delay line takes a bunch of samples and looks at a waveform. There are certain numbers of levels it can remember, and if the amplitude sample falls between two of those levels the A/D converter averages and makes a decision. As an example, when the signal is bigger than level 500 and smaller than level 501 but it's closer to level 500, the converter rounds it off to level 500 and that's transferred down and later converted back to audio. Now if the signal happens to be exactly 500 that's great, but if it happens to be 500.4 then there's an error. Note that it had to make an error; it didn't make a mistake like dropping a bit or something, it made an error it's designed to make. Now, if you have 16 bits of computer information defining the amplitude of a signal you have much smaller increments, so it can be much more accurate...which is part of why 16 bit lines are in the majority.

To sum up, a signal is averaged up to a bunch of numbers; these numbers are transferred down the memory bank and then converted back to audio again. This rounding off is called quantizing. Let's say you have 15,000 samples of a 1 KHz sine wave; that's 15,000 samples per second, and you're remembering 15,000 instantaneous amplitudes. If, after being delayed you put all those samples back together in the right order you get your sine wave back again. The fact that the analog to digital converter rounds off, though, means that it makes errors and this is what quantizing error is all about. Digital delay lines have this error, you can hear it, and this is one of the big problems with digital delay.

"WE LIVE IN AN ANALOG WORLD, A WORLD THAT DISTORTS THINGS IN A FASHION THAT MAKES SENSE...THEN CAME DIGITAL WITH DISTORTION PRODUCTS THAT DON'T MAKE SENSE, AND WITH REVERSE DISTORTION LAWS, SO IT'S NO WONDER DIGITAL SOUNDS JUST PLAIN WRONG TO A LOT OF PEOPLE."

P: What does this error sound like?

S: This is of course very subjective; sometimes, it's a very annoying "gricky" sound. People who have used digital delay for a long time realize that you get a gritty kind of coarse sound. If you operate a digital line at 0 VU with a full signal sweep, you're using the majority of all those little samples and things aren't too terrible. This will give you usable distortion levels at full amplitude levels. But as the amplitude goes down in digital delay lines, the distortion goes up...a very unnatural effect. The reason the distortion level goes up is because you have your signal excursions over less of a step so each amplitude step becomes coarser and coarser until at the end, in many digital delay lines you can actually hear it going (makes sound similar to that of a sputtering Woody Woodpecker) as it quantizes the relatively giant steps towards the end. There's also quantizing noise, which is the switching transient noise that's transferred down, and little errors at the bottom that we hear as kind of a grit. What happens in a digital delay line is as you approach lower levels you literally approach 100% distortion...and, this distortion is not particularly pleasing. When you splatter a digital delay line you don't get harmonic distortion; you get bastard harmonics, harmonics that are the product of the input information's harmonics plus the clock frequency. For example, if you're sampling at 40 KHz and you splatter a 5 KHz tone, you get a 10 and a 15 KHz harmonic...but it's not just that simple. You get 10 and 15 KHz multiplied and divided by 40 KHz, the clock rate, which gives you a totally unnatural series of sideband harmonics. Now, you can accumulate a great deal of "natural", harmonically related distortion and it will sound acceptable, since your ear has grown up with harmonically related distortion. Your ear has also grown up with another law: the louder it is, the more distortion...whether it be your ears distorting or the equipment. For example,

if I play a record and it's distorting 5%, you'd go "that's really loud". So if I say "OK, I'll turn it down", then go over and turn a knob that takes only the distortion out you'd probably say "that's much better". You'd have to be very well trained to realize that I have not done anything except turned down the distortion...you correlate distortion that much with amplitude. We live in an analog world, a world that distorts things in a fashion that makes sense. Wellll...then came digital with distortion products that don't make sense, and with reverse distortion laws, so it's no wonder digital sounds just plain wrong to a lot of people. Now, with the new PCM tape recording systems and people going to 16 bits and doing floating point conversion it's getting fancy enough so that all these awful things we just talked about have to be weighed against the inherent awful qualities of tape...but that's a whole other subject.

There's also delta digital. Delta is kind of a cheating system where everything is put through in serial; information is not converted to a number of parallel digital bits. There's one delay line and all the information is spit out in serial fashion. I won't go into Delta too much except to say that with the present technology it's extremely difficult to push Delta to anything that even resembles professional quality. Delta lines can't slew, and that's the key: you can't get things to happen fast enough to make delta work under true professional conditions. There are some advantages to the manufacturer with delta, though; for example, it's a snap to convert from digital to analog in delta. Since delta is a series of numbers formatted so that they come out as steps of amplitude you hang a smoothing filter on the output and bang---you've got audio. This is great for the manufacturer because it's super cheap to make. But it's still digital and it's still subject to digital error, along with some of the worst characteristics of analog...high frequency problems and the like. OK, now I've slammed around some of the competition, on to the analog delay lines. There are two types of analog lines: bucket brigade and a new type of CCDs...

P: CCDs?

S: Charge coupled devices, bucket brigades being BBDs. BBDs are pretty simple to use; they can also be swept over a pretty wide range. But frankly, the performance disadvantages are so severe, they cause many, many problems...BBD delay lines are extremely noisy; all kinds of noise gating and companding concepts have been designed to try and quiet them down, but they're still noisy. It's also unrealistic to try and achieve very long delays with bucket technology. And, as you increase delay you lose bandwidth; i.e. the longer the delay, the more the high frequency rolloff. More disadvantages: as you increase delay BBDs become noisier and I'm not talking about something subtle, I'm talking about becoming virtually unusable. They're also temperature sensitive. Another problem shows up when you try to develop a BBD product: quality control. You can buy a hundred chips and you get five different types of performance parameters...they're just not that tightly controlled. In spite of all these problems, there are quite a few bucket brigade analog delay lines out. Almost every analog delay line on the market uses some form of bucket brigade and they're all subject to the same problems I've just mentioned. But, there is a new type of analog transfer system and it's CCD. A well designed CCD element, used in a well designed delay line format, can be free of these problems. The Marshall Time Modulator and our other products use this new system.

(to be continued)

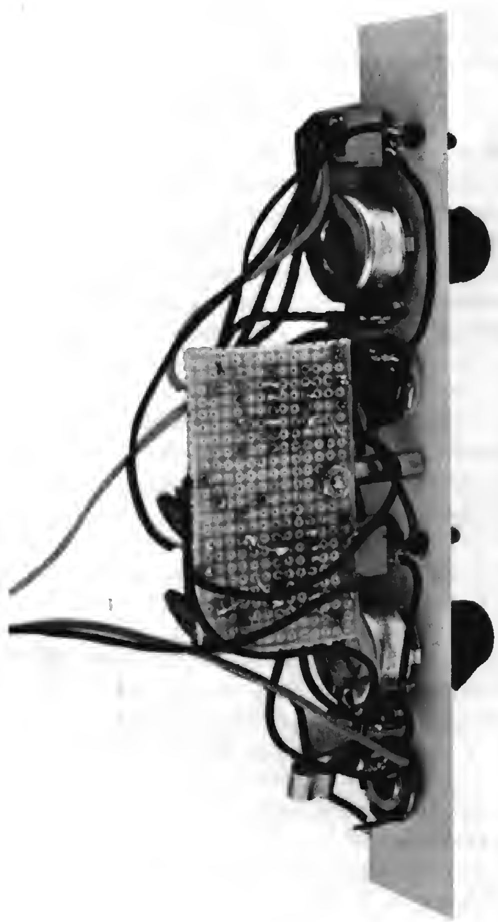
(AMS-100 cont. from page 1)
on plexiglass panels that mount into a wooden frame. The frame is approximately 19" wide so that it is about the same width as a standard rack panel.

I realized as I got into building the AMS 100 that I would need to go through two packaging stages. In stage 1, I would be able to build up all the modules and try them out with each other in various combinations. However, since I might want to change a module, make revisions in the light of reader suggestions, and just generally need a certain amount of flexibility, I opted for a packaging scheme that would be functional but not permanent. In stage 2, after finalizing the layout and number of modules, I could then repackage them into a standard rack mount panel.

To simplify construction during stage 1, I built the circuits on plexiglass panels because plexiglass is easily drilled (with the proper bits), colorful, and you can get scrap plexiglass inexpensively from a distributor. Figure 2 shows a typical module (the dual envelope generators). Note that the circuit board is *(Cont. on page 12)*

(AMS-100 cont.) raised above the panel by a spacer to allow room for other components underneath it. The three wires you see trailing off into the distance lead to the power supply; the trimpots face the back of the module for easy calibration. Remember if you do decide to use plexiglass to make sure you connect the pot and switch cases to ground to prevent hum and noise pickup.

The various panels are about 7 inches tall, with another ½" of overlap on both top and bottom to allow it to screw securely into the wooden frame. If you'd like your



modules to be size-compatible with Moog, ARP, or Aries modular systems, you will need 9" tall panels.

Repackaging the modules for stage 2 is not too difficult. First, you select a metal rack panel that's either 7 or 9 inches tall, depending on which size you're using. Then, you carefully unscrew all the screws, pot mounting hardware, jacks, and switches from the plexiglass panel... do this for each module. Next, lay the module front panels over the metal rack panel and use them as templates for drilling holes. Note that during the repackaging process you might want to add some extra jacks or controls; no problem, just leave a little space between the modules for inserting these new parts. Drill out the panel, then re-mount the modules into the metal rack panel.

POWER SUPPLY CONSIDERATIONS. Several people have asked how much current capacity is required for an AMS-100 system. Since this is a modular system, it's impossible to predict the current demands...a system with 14 filters is going to draw more current than a system with 2 filters. However, 1 Amp per side should hold you over for at least a good while. These modules don't draw a whole lot of power, so you can power quite a few modules before the supply poops out. I'll be giving specs on these modules, including power supply consumption figures, in a future issue.

I've elected not to run a schematic of a ±15V supply, since I have the feeling that most DEVICE readers will have the knowledge to put one together themselves. If you need a reference, though, I covered a suitable supply in my Home Recording for Musicians book; I also covered a ±9V supply in the December '78 issue of Guitar Player. The ±9V project may be converted to ±15V operation by doing the following:
1) change the transformer from 24 VAC center tapped to 30 VAC center tapped 2) replace the 7808 regulator with a 7815 type 3) replace the 7908 regulator with a 7915 type 4) short out the two 1N4001 diodes going from the ground terminals of the regulators to the common (or ground) line 5) change the output capacitors to units with a 15 VDC or greater working voltage.

Since Godbout Electronics supplies a parts kit for the ±9V power supply described in Guitar Player, I modified it as described above and now use that power supply to power the various modules. Figure 3 shows the power supply board with its modifications.

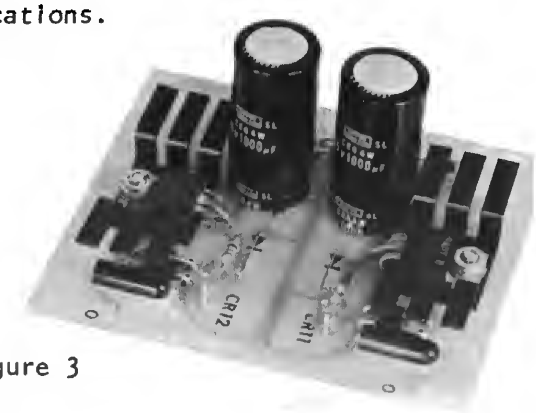


Figure 3

DISTRIBUTING POWER TO THE MODULES. Best wiring practice involves running separate wires from the positive and negative power supply terminals to the positive and negative supply points on each module (see figure 4). In addition, a separate ground wire should run from the power supply to the ground point on each module. If the power supply is located at some distance from the modules, then also include a couple of 100 uF 15 VDC capacitors across the supply lines as shown.

With plexiglass panels, you'll probably end up running the ground connections on the jacks to the ground point on the module. With a metal panel, I'd suggest NOT connecting the jack grounds to the module, but rather, let the jack grounds make contact with the panel and then ground the entire panel to the power supply ground point as shown in figure 4. This will minimize any tendency towards ground-looping.

I'd like to add that power supply distribution is not trivial, so follow these guidelines closely...running supply lines haphazardly from module to module is not as good as running separate lines to each module from the power supply.

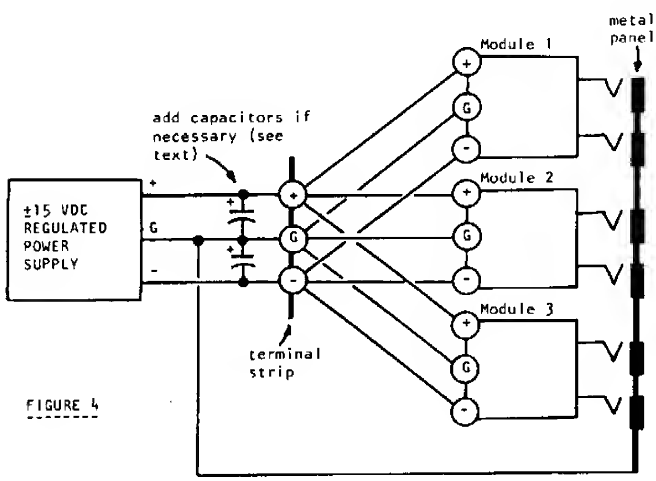


FIGURE 4

Note: I tried substituting LF356 ICs for the 741s originally specified for the envelope generators, and they gave much better performance since they loaded down the timing capacitor so very little. Another mod is to use a faster quad analog switch, like the 4116 or 4066, instead of the slower 4016. This improves the attack characteristics somewhat. Now you see why I use sockets all the time!

OK...next month we'll cover a high performance VCA built around the 570/571. The month after that, we'll look into VCFs.

(Pedalboards cont. from page 7)

seem like an obvious thought, but there are plenty of projects I've seen in which the ease of operation seemed to be the last thing on the designer's mind. Once you have identified these initial needs and a design starts to form in the mind's eye, commit it to paper. Start with a list of your present effects and determine the things that please you and those that annoy. On a separate sheet give yourself a preliminary sketch of how you want your system to look and take into account, wherever possible, that you want to emphasize the positive aspects of your "toys". Take measurements of the boxes your effects come in (if you've decided to leave them intact) and take these into account in your design. Leave room and flexibility for the addition of future devices - you'll be glad you did. Use your own body as the basis for determining measurements and positioning. Size of arms, hands, and fingers are important factors when it comes to actual use. Get a tape measure or ruler and write this data down. Here's an example of the importance of this aspect of design: I once helped develop a footswitch board, but in our rush to get the thing into production we neglected to take into account the concept of a fulcrum. There were plenty of switches to mount and we jammed them on to a 19" wide board that sat upon little rubber feet. Unfortunately, one of the switches was about 1" from the end so that every time you stepped on the switch...FWAPP!!...the board would hit you in the leg. If we had put balance into the design in the first place, this unpleasant problem wouldn't have happened. Allow yourself to live with your design, realizing at the same time that as you progress things will change. If you or a friend can do drafting, turn your sketch into something a little more definite. This will help you to visualize the project even better.

The next step in research (actually this should be going hand in hand) is to gather information related to your project. This will mean, for some of us, picking up books on basic electronics, soldering, and shop techniques. These are usually available at the public library - emphasize those words, PUBLIC LIBRARY - or if you wish to invest, visit a good book store or electronics shop (see bibliography for titles). The knowledge of shop tool use is important because

(Cont. on page 14)

(Pedalboards cont. from page 13)
(Pedalboards cont. from page 13)

we will be using a variety of materials - wood, metal, and plastics. Proper use of tools and materials will prevent us from re-inventing the wheel. Other information of use will be schematics of the various devices you own; but obtaining these is usually not a simple matter. Manufacturers are frequently not inclined to offer you access to their "secrets", so you may have to do without. Acquire data on hardware you feel you will be using in your system (knobs, switches, pots, cable, jacks, plastics, plywood, etc.) and FILE IT FOR REFERENCE! You are building an organizing system for effects, so attack the process in an organized fashion. For leads on this kind of information pick up copies of magazines like Popular Electronics and Radio-Electronics and read the ads. One of the greatest sources of information at your disposal is the yellow pages; in most cases that's the first place I go, since it's close at hand and I can usually find out what I want to know through a phone call. Take note: when dealing with those whose knowledge may exceed yours in certain areas it is best to come as prepared as possible. When dealing on the phone or in person with a particular problem, write out what you need to know ahead of time as clearly as possible. Take notes of the reply. When trying to gain information be professional! You needn't give the impression of a technician, but at least come off as someone who knows where they want to go. The folks in your friendly neighborhood electronics shop and hardware store can be of great help in your search for knowledge...they can also be of no help at all, but give them a chance. It often takes time to build up a rapport with sales people, but the benefits can make it more than worth your while. A great source of information, of course, is a subscription to DEVICE (obvious plug!).

While this is all going on, keep refining and visualizing the ultimate outcome of your efforts - a completed functional effects system. Future pedalboard articles will concentrate on specific aspects of putting your system together.

At this point I'd like to address myself to those for whom this will all seem pretty basic; if you know more than I do, your input to this column is once again solicited. I want to offer your solutions to problems encountered in building pedalboards to DEVICE readers. Interaction is the word!

BIBLIOGRAPHY

The following are sources of information that will be helpful for our future undertakings. We will update this from time to time based on your suggestions.

UNDERSTANDING SOLID-STATE ELECTRONICS
Texas Instruments Inc., Mail Station 84, PO Box 5012, Dallas, TX 75222. An easy to read and easy to understand treatise on basic semiconductor theory. $2.95

HAYDEN ELECTRONICS 1-7 SERIES
Hayden Book Co., Rochelle Park, NJ. More accessible basics, often available at book stores for about $5 a book. Available individually or as a set.

ELECTRONIC PROJECTS FOR MUSICIANS by Craig Anderton. Music Sales Corporation, 33 West 60th Street, New York, NY 10023. If you don't have this one, what are you doing here? Costs $7.95.

COMPLETE BOOK OF WOODWORKING by Rosario Capotosto. A Popular Science book club selection. PO Box 2006, Latham, NY 12111. One of my favorites on the subject. If you flunked shop or would just like a good wood workers "bible" this is it. You're gonna need it! Write the book club for price.

MUSICAL ENGINEER'S HANDBOOK by Bernie Hutchins, or more properly, edited by Bernie Hutchins. Electronotes, 1 Pheasant Lane, Ithaca, NY 14850. More technical than previous listings but a good source for the more advanced. $18 pages only, 3-hole punched; $20 soft cover binder.

INFO: ***Paul Rivera,*** *pedalboard specialist and formerly with Valley Arts Guitar Center, now offers a custom pedalboard service along with Fender/Music Man amp modifications and hardwood replacement amp cabinets for same. Paul is one of the finer people in this business whose work is well-known in LA's professional circles; DEVICE wishes him well. For further info send self-addressed stamped envelope to PO Box 641, Tujunga, Los Angeles, CA 91042 or call (213) 352-4800...The big thing in electronics the past few years has been the microprocessor. Now, in a trend that bodes well for audio experimenters,* ***small microprocessors are being produced with integral A/D and D/A conversion.*** *This should make it easy to digitally process analog signals, leading to products like very inexpensive digital echo and reverberation.*

DIALOGUE

We'd like to thank you for your comments and suggestions, as they help us to create a publication that truly answers your needs. Send ideas, love letters, crank mail, or whatever to DIALOGUE, c/o DEVICE.

Dear DEVICE,

We wish to thank you for the very objective review of the IVP that appeared in Vol 1:2-79. A few additional points may be of interest to DEVICE readers...

All IVP units shipped since December, 1978 have individual IC sockets installed to facilitate servicing. We heartily agree that this is a most convenient improvement.

Your suggestions for the use of higher quality pots, the inclusion of additional features such as pre-eq/post-eq effects loop switching, etc. are all very valid and useful improvements. It should be recognized, however, that all such improvements may not be universally desired and not every musician may wish to invest in that degree of sophistication. Should we receive noticeable requests for a more sophisticated unit, we would be happy to produce such a device.

As you noticed, there exists a very wide range of input/output levels and impedances in the effects devices being produced today. The IVP was designed to provide workable interfacing with as many devices as possible. There appears to be a need for all manufacturers in our industry to reach for a level of standardization in these specifications. This would greatly benefit not only fellow musicians, but peer manufacturers as well. We at INTERSOUND strongly support this concept.

The staff of DEVICE is to be congratulated for providing authoritative product reviews, meaningful application ideas, and just plain good information. We are sure that many musicians have been anxiously waiting for such a publication. We wish you the best of luck.

Ray Wilkinson
VP Marketing
Intersound, Inc.
Boulder, Colorado

Dear DEVICE,

Kudos on AMS-100. Will circuit boards be available from Godbout or other sources? I really rely on these as opposed to scratch building. If not, can you include board layouts?

By the way, please don't give up your GP column - you could put the complex stuff in DEVICE, the simpler stuff or advice in GP.

Art Josselyn
Southampton, PA

Art - Godbout's is checking into the feasibility of handling AMS-100 boards; also, DEVICE reader Rick Norman is laying out AMS-100 boards for his own use, and I have approached him about possibly making his artwork available to other readers for a nominal charge. So one way or another, we'll probably have at least artwork pretty soon.

As far as Guitar Player is concerned, I love writing the column and will continue to present a blend of construction/tips, simple/complex material in GP's pages. DEVICE is designed to handle projects that are too lengthy or esoteric for a general interest magazine like GP; I believe there is a need for all these types of articles. ---Craig

Dear DEVICE,

This magazine is a dream come true. I would like to thank the people who made it possible.

Suggestions for articles: 1) a full length in depth article on single and double coil pickups 2) a review of MXR's digital delay 3) info on pots with push-pull switches.

Tom Brown
Salida, CO

Tom - Thanks for the kind words. Re suggestion 1, see the Tele review in this issue; it should answer many of your questions. 2: We'd like to publish a review of the MXR Digital Delay. Are there any readers out there who'd care to tackle this? If not, perhaps we could get MXR to loan us one for evaluation. For 3, Schecter Guitar Research (PO Box 9783, North Hollywood, CA 91609) carries high quality pots that include push-pull switches. You can wire them up as DPDT, DPST, etc. Contact Schecter for prices and ordering info.

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